VOLUME II NUMBER 4 DECEMBER 1986 PRICE 3.50

EXPERIMENTAL MUSICAL INSTRUMENTS

FOR THE DESIGN, CONSTRUCTION AND ENJOYMENT OF NEW SOUND SOURCES

THE TAPE IS HERE!

WE ARE PROUD TO ANNOUNCE that the much discussed, long anticipated, eagerly awaited, painstakingly assembled, finely crafted EMI tape has been completed and is available. It is a forty-five minute cassette called **From the Pages of Experimental Musical Instruments.** This first EMI tape contains music of instruments which appeared in **Experimental Musical Instruments** Volume I; that is, in the issues of EMI's first year of publication, June 1985 through April 1986. The work of ten builders playing one or several instruments each is represented:

Ellen Fullman plays the Long String Instrument;
Robert Rutman and friends play steel cello and bow chimes;
Pierre-Jean Croset plays the 18-string Lyra;
the four members of the Glass Orchestra play a million and one instruments of glass;
Susan Rawcliffe plays several odd and exotic fipple flutes;
Wendy Chambers plays the Car Horn Organ;
Tom Nunn plays Mothra and Crustacean;
Sharon Rowell plays triple and quadruple ocarinas;
Bart Hopkin plays Disorderly Tumbling Forth;
and the wind plays the Puget Sound Wind Harp built by Ron Konzak.

IN THIS ISSUE

The ten contributors have provided us with a set of wonderfully diverse performances -- fascinating as documentation of the instruments, and, more important, satisfying from a purely musical point of view as well.

Included with the cassette is a tiny sixteen page booklet, made to fit (believe it or not) in the cassette case, with information on the instruments played. More information on the instruments is available, of course, in the issues of EMI in which they appeared.

The tapes are available to EMI subscribers for $6 apiece, and to non-subscribers for $8.50. Checks should be made out to **Experimental Musical Instruments**, P.O. Box 784, Nicasio CA 94946. For convenience, subscribers will find an order form in the envelope along with this issue.

A second tape corresponding to EMI's second year will appear when Volume II of the written journal is complete, in June 1987. We already have much of the material for that second tape in hand, and, let me assure you, it looks to be as fascinating as the first.

NOTICES

NEW INSTRUMENTS / NEW MUSIC: RICHARD WATERS LAST MONTH; TOM NUNN SOLO NEXT

The latest concert in the New Instruments / New Music series was presented by Richard Waters, playing the Waterphone (featured in EMI's last issue). He performed "Non-Composition for Three Waterphones," accompanied by slides of his watercolors and photographed landscapes. In the context provided by the slides, oriented as they were to the natural beauty of the world around us, the sound of the Waterphone was freed of the irrelevant extraterrestrial program that is often associated with it. It was there simply as music, and it came across expansively and richly. Richard also showed slides of some of his other instruments and sculptures, discussed his work and answered questions. All in all, a fine presentation.

The next concert in the series will be presented by Tom Nunn, series organizer, playing solo on space plates, electroacoustic percussion boards, and who knows what else. Should be very good.

The concerts take place at 3016 25th St., San Francisco, CA 94110, at 2:00 pm on the first Sunday of odd numbered months. For the Tom Nunn concert, that's January 4th. Admission is $5. For more information call (415) 282-1562.

CHILDREN'S CONCERT -- Short Sound Stories. Sunday, 12/14/86, 11:30 at Los Angeles Municipal Art Gallery, Barnsdale Park; with Alex Cline and Ron George, percussion.
CONCERT -- Sound Stories, Thursday 12/18/86, 7:30 pm, $5 donation; with Alex Cline, Ron George, percussion and Jim Grant, vocals. Program: "Mocking Bird & the Bear," a Grimm Brothers fable set to music, "Ancient Beasts" didjeridu dance, a timbral piece with slides & lights, flute and flute & percussion works.

THE EMI TAPE IS HERE AND AVAILABLE! From the Pages of Experimental Musical Instruments Volume I includes music from instruments featured during EMI's first year of publication. The price is $6 for subscribers; $8.50 for non-subscribers (postage, etc. included), from Experimental Musical Instruments, P.O. Box 784, Nicasio, CA 94946. More information is on the front page of this issue.

EXPERIMENTAL MUSICAL INSTRUMENTS
Newsletter for the Design, Construction and Enjoyment of New Sound Sources

ISSN 0883-0754

Editor
Bart Hopkin

Editorial Board
Prof. Donald Hall
Roger Hoffmann
Jonathan Glasier
Jon Scoville

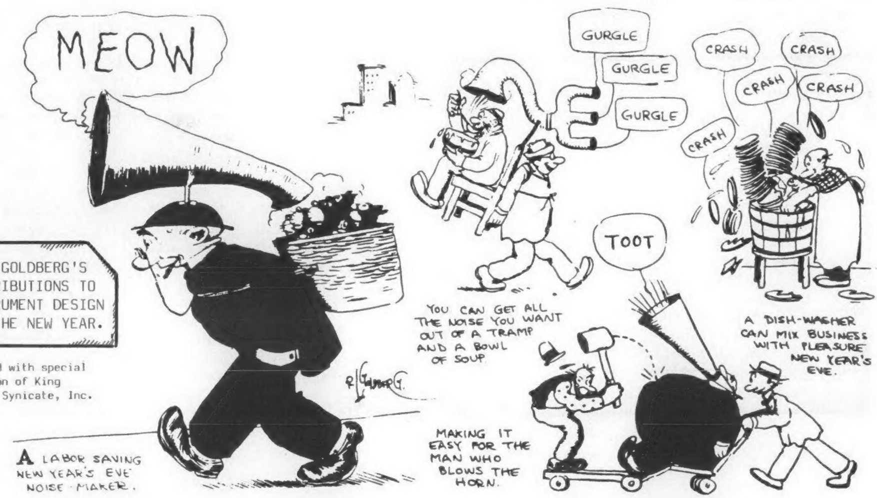
Published in February, April, June, August, September and December by
Experimental Musical Instruments
P.O. Box 784
Nicasio, CA 94946
(415) 662-2182

Subscriptions $20/year ($27 outside the U.S., Mexico and Canada).
Back issues $3.50 apiece.

ADVERTISING POLICY: Experimental Musical Instruments is set up to survive on subscriptions rather than advertising, but we will run ads which are potentially interesting or valuable to the readership. Please write for advertising rates.

Subscribers can place relevant classified ads of up to 40 words without charge, and they will receive a 15% discount on display ads.

SUBMISSIONS: We welcome submissions of articles relating to new instruments. Articles about one's own work are especially appropriate. A query letter or phone call is suggested before sending articles. Include a return envelope with submissions.



RUBE GOLDBERG'S CONTRIBUTIONS TO INSTRUMENT DESIGN FOR THE NEW YEAR.

Reprinted with special permission of King Features Synicate, Inc.

POLYCHORD 1
AND MICROTONAL STEEL GUITAR FRETBOARDS
by Siemen Terpstra

Siemen Terpstra is a microtonalist, theoretician and instrument builder living in British Columbia. In the following article he describes his system of fretboard markings for steel guitars (that is, guitars designed to be played using a sliding metal bar on the strings rather than pressing them against the fretboard). Siemen's fingerboard overlays serve as guides to placement of the bar as well as conceptual organizers of pitch relationships perceived either musically or mathematically. They also happen to be very beautiful.

Siemen Terpstra can be reached at 4472 James St. Vancouver, B.C., Canada, V5Z 3J1; (604) 879-7008.

Some years ago I had the opportunity to visit Ivor Darreg when he lived in Glendale. There I played his Megalyra, and became fascinated in the potential of the steel guitar as an "experimental" instrument. [For an article on Ivor Darreg's Megalyra family of instruments, see EMI Vol. II #2.] On coming back home to Canada, I began to alter an old Gibson lap-steel, but was frustrated by its short scale and its small number of strings. I began to dream about the "ultimate" steel guitar for my musical purposes. Such an instrument would have a fairly long scale, though not as long as the Megalyra, and many more strings than the Gibson steel.

The dream became an actuality with the building of Polychord 1. I had developed a number of designs for new musical resources, and was fortunate to receive funding from the Canada Council to put together Polychord 1 as well as the Modular Keyboard, a new approach to Tuning Utilities Software for Apple Computer Music Systems.

Polychord 1 is a large steel guitar with two "necks" of fourteen strings each. The active string length is 100 centimeters, which is much longer than a standard guitar (66.5 cm), but short enough so that the stretch is not unmanageable in playing it. The wood is laminated cherry and African walnut. The bridges are brass. I have used electric guitar pickups which are wired in series and in parallel with toggle switches to choose the pickup configuration.

So far, this arrangement does not seem to be overly unusual, but a glance at the photograph reveals a complex, multicolored fretboard. In fact, I have made many fretboards for this steel guitar. Since the fretboard is really only a reference guide for the sliding bar, it is possible to replace one fretboard with another if one is interested in working in a different tuning system as a musical reference.

The steel guitar is closely related to the traditional monochord. We have simply replaced the movable bridge with the sliding bar, and, of course, added pickups. One of the reasons that I built this instrument was to have an ideal tool for displaying principles of musical acoustics and tuning theory. The traditional monochord was calibrated with fretlines marked in numerical ratios. I have replaced the ratio numbers with colored patches in order to make the fretboard more practical for actually playing. The fretboard may be calibrated for any tuning system that we wish to use. It is wonderfully open-ended.

I have been interested mostly in equally tempered intonations since they offer unlimited modulation. My favorites are 12, 19, 31, 43 and 53 divisions to the octave. For the purposes of this short article, I am focusing on 31, but the same principles apply for any division. Also, any steel guitar, including the commercially available varieties, may be outfitted with home-made microtonal fretboards of the type described here.

The remainder of this article is a brief "how-to" on making practical fretboards for the steel guitar of your choice. It is really not too difficult, and the potential for the exploration of alternative harmony is vast.

To begin, measure the active (bridge to bridge) string length of your instrument. Polychord 1 has a fairly long scale at 100cm, partly because I wanted to apply the 53 tone equally tempered division to it. (53-E.T. is a good mimic of Just Intonation, so that I can demonstrate traditional Hindu, Chinese, and Greek scales with sufficient accuracy.) With this string length the frets are not overly close together, even with 53 to the octave. But any string length will do. Most commercial steel guitars have a string length similar to a standard guitar, and such a length is useful for divisions up to 31. Now that you have the string length you can plug it into a formula that gives you the proper fret positions.

We will draw our fretlines on a piece of white poster-board or cardboard which has been cut to fit over the standard fretboard of the given instrument. Later, when the fretboard is finished, we will laminate it so that it can be cleaned and protected from smudging, etc.

The calculation and drawing of the fretlines is

SIEMEN TERPSTRA
PLAYING
POLYCHORD 1

the crucial part in making our fretboard, so we try to be as accurate as we can. Hence, I must get a little technical here. Don't worry though, you won't need a Ph.D. to follow this, it's just a little arithmetic. We must use a series of numbers called a **fret table**. Every tuning system has its own fret table, and you can generate these numbers yourself. For equal tempered scales this can be done using a log table and a pocket calculator. Alternatively, you can acquire the information from a number of sources. I am one of those sources. Just write me for the info if you are stuck. The fret table is just a series of ratios which are peculiar to **that** particular tuning system -- ratios which define the size of the interval steps.

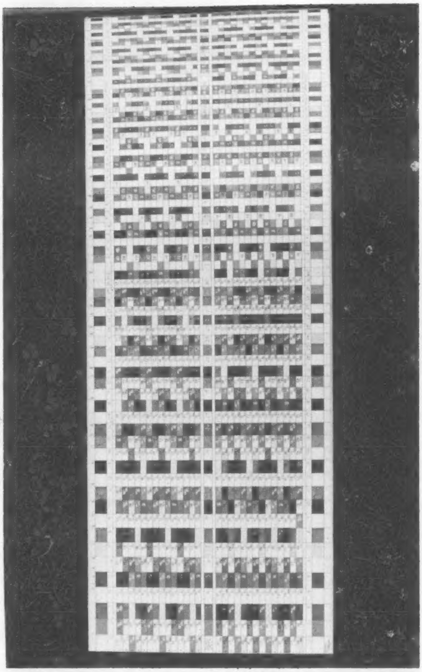
Briefly, here is a the procedure for generating the fret table for 31-E.T. We must calculate the thirty-first root of two, which can be done with the aid of a logarithm table. To save you the trouble, the answer is approximately 1.02266643. This number is used to find the position of the first fret. Now take this number and successively multiply it by itself (a calculator helps!) to find the "magic numbers" for the other frets. [For more on the reasoning behind all this, see "Scales and their Mathematical Factors" in **EMI** Vol I #5.]

I am reluctant to write out the complete fret table for 31-E.T. in this article, since I do not wish to take up so much space with a long list of numbers. However, I will give the first few so that you see how they are used. For 31-E.T., the first five numbers (out of 31 altogether) of the fret table are:

0--1.00
1--1.02266643
2--1.04573415
3--1.06937970
4--1.09355991

These numbers will give us the proper fret position for **any** string length. The rule is this: Divide the string length by the appropriate fret table number to get the distance from the bridge to the given fret. For example, say that our string length is 100cm. Then the first fret is 100/1.02266643 = 97.783597cm from the bridge, which is 100 - 97.783597 = 2.21641cm from the "nut" (the other bridge). In actually measuring it, we would round off the number to 2.22cm, and draw the fretline parallel to the nut. As a second example, suppose our string is 24 inches long. Then the third fret is 24/1.0693797 = 22.442917 inches from the bridge, or 24 - 22.442917 = 1.557083 inches from the nut. Obviously, it is easier to work in the metric system here, rather than the British system.

In a like manner, we must measure the position for each fret and draw it on the fretboard. All this is rather tedious, but when it's done, the rest of our job is easy and fun. On my fretboard, I have calibrated the frets for two octaves and a fifth, which makes a grand total of seventy calculations. But the second octave is easy. Just divide the first octave position by two. It is fairly easy to make the accuracy good to about a millimeter.



SIEMEN TERPSTRA'S FINGERBOARD OVERLY FOR POLYCHORD 1 -- but it's a shame we can't show it in color, for it really is a lovely sight.

We now have a fretboard covered with a lot of parallel lines which get closer together in a logarithmic sequence as we move up toward the bridge. The next step is to convert this board into a bunch of rectangular patches by filling in the lines parallel to the string lengths. Measure the distances between your strings as they cross the nut and bridge, then connect the lines so that each string has a patch under it for each fret position. Now the fretboard looks like some weird, alien chessboard!

We are almost ready to apply the color code and tint the patches -- an essential step in making the fretboard practical. But first, we must decide how we are going to tune the strings. This decision depends upon our musical purposes. Polychord 1 has two banks of strings, so that I can have alternative patterns. The first bank I have tuned (from low to high pitch) A-C-E-G-A-B-C-E-G-A-B-C-E-G. This tuning gives me a major triad, major six chord, major seven chord, minor triad, minor seven chord, minor flat-six chord, a couple of nine chords, and a suspended triad. The second bank of strings is tuned (from low to high pitch) A-B-D-F-G-A-B-D-F-G-A-B-D-F. This tuning gives me a dominant seven nine chord, and a minor sixth eleven chord. Of course, you do not need to copy this tuning. Perhaps the tuning you want is C-E-G-C-E-G-C, if your steel guitar has only seven

strings. The steel guitar being such an open-ended instrument, it seems that each player prefers a different tuning set-up. That's great, but remember that whenever we use a different tuning pattern, we must build a separate fretboard for that pattern. So it is good to have this decision made beforehand.

We are now ready to number the fret patches. The 31 tones per octave of this tuning system are numbered from 0 to 30, starting with C as 0 (by convention). For example, the pitch A has number code 23. Therefore, if one of our strings is tuned to A, then the patch under the first fret will be number 24, and so on. Alternatively, we can write the actual pitch name on the patch instead of the number, or both for that matter. The reason we do this is because I am using a sort of "paint by number" approach to coloring fretboards. The number and/or pitch name also help us find our way when we play.

With this numbering done, we can apply the color code to the fretboard. Before laying our the table though, I shall briefly explain the rationale behind the color code, which is not arbitrary. I wanted a color scheme which would highlight about half of the notes -- those notes in closest harmonic relationship to our arbitrarily chosen reference tuning pitch (C). With this arrangement, the most important frets would be emphasized in the same way that the fifth and seventh frets of a standard guitar are emphasized by an inlay.

The natural solution to this problem is to apply the rainbow sequence (red, orange, etc.) to the set of modal relationships arising from the harmonic series and its reciprocal. Thus the musical fifth and fourth are red, the major third and minor sixth are orange, the minor third and major sixth are yellow, the whole tone and minor seventh are blue, and the tritones are purple. This overall system may be applied to many different tuning systems. In the case of 31-E.T., a further refinement is necessary. For example, there are two sizes of major thirds, a "just" form and a "septimal" form. Therefore I use two shades of orange. The scheme is symmetrical about the generator tone C. This tone should also have a color, which should be white or "shining." Consequently, I use a mylar strip for C, an effect which enhances the beauty of the fretboard. Here is the color code for 31-E.T.:

FRET NUMBER		PITCH NAME		COLOR
0=31		C		Mylar (shining)
2	29	C#	Cb	Light Blue
3	28	Db	B	Deep Blue
5	26	D	Bb	Green
7	24	D#	Bbb	Light Yellow
8	23	Eb	A	Deep Yellow
10	21	E	Ab	Deep Orange
11	20	Fb	G#	Light Orange
13	18	F	G	Red
15		F#		Light Purple
16			Gb	Deep Purple

The other frets are left uncolored, to good effect. The result is a patchwork quilt which enables the player to find the right fret with far greater ease than the old monochrome approach. However, keep this limitation of the color code in mind. Different microtonal tuning systems require different variations on this general approach. I have given you the best scheme for 31-E.T., but the best color code for, say, 19-E.T. is a bit different. To elaborate these differences and the reasons for them is beyond the scope of this short article. You could write me for details on the system of your choice. I even have a color scheme for my 12-E.T. fretboard which is a distinct improvement over the old approach. Besides making playing easier, it also looks attractive!

I filled in the colors using felt-tip pens, but you could use acrylics or various other media. It is also handy to have an extra fret patch alongside the strings on both sides as a general guide. This patch gives the general fret number so that we can use a tablature system in notating the music -- a notation system peculiarly suited to steel guitars.

A word about tuning. When using this 31-E.T. fretboard it is advisable (though not strictly necessary!) to set the temperament accordingly. However, I have found that I can tune my strings justly and still use this fretboard if I use vibrato or other means to mast the inaccuracy. Also, since the 31 system is sonically indistinguishable from one-quarter comma meantone tuning, we can apply a meantone tuning regime to the strings. This is not the appropriate place to describe how to tune meantone, but this tuning is much easier to implement accurately by ear (with the use of appropriate harmonics) than is 12-E.T. I have developed a program for my computer which lets me use the terminal as an "electronic tuning fork," so that I can set the temperament for whatever system I care to investigate. Another approach, which is not as accurate, but inherent in the instrument, is to use the fret itself as the tuning guide, much like an ordinary guitar. It's low-tech and simple. Don't try to use those electronic tuners, though. They are all stuck on 12-E.T. When will the music industry finally come up with an electronic tuner which is inter-system? Any company trying to develop such a tuner should consult me for design considerations!

Polychord 1 does not have foot pedals. The reasons are various. Since I am interested in microtonal systems, I would need foot pedals which could produce different sizes of semitones. I concluded that such engineering is not worth the effort. Moreover, in my experience I have found that foot pedals are not all too accurate. The player can adjust for the inaccuracy by shifting the bar, but this procedure lowers the overall tuning accuracy. However, if your present steel guitar does have pedals, you can still apply a microtonal fretboard to the instrument.

The steel guitar has long been "ghetto-ized" in its association with country music. My musical bias has been more in the direction of Asian or intercultural approaches. Clearly, the instrument is amenable to a host of different sonic directions. I hope that this article encourages the use of the instrument for the exploration of new harmonic resources.

With this article, as it has once before, EMI breaks from the narrower focus on new instruments to look at a lesser known older instrument which may be of special interest to new instruments people.

THE GLASS HARMONICA
Originally designed by Benjamin Franklin
Currently being built by Gerhard Finkenbeiner

Article by Vera Meyer

People have been tapping and striking glass objects as a means of making music since Persian and Chinese antiquity. It wasn't until the mid-1700s in Europe however that history records the first occurrences of music made by rubbing the rims of glass goblets with moistened fingers to produce various notes. The goblets or wine glasses were firmly affixed to a table, each precisely tuned with water to a different pitch. The musician would sit in front of this assembly, making the glasses sing as he moved a wet finger steadily and with a light pressure around the rims. This arrangement was called the "musical glasses" and for a time was quite the rage of high society in Europe.

When Benjamin Franklin was in Europe as Ambassador to the Colonies he chanced to hear a concert played on a set of fifty such tuned water glasses. Being charmed and captivated by the beautiful sounds emitted from the glass, he immediately set out to design a more convenient and practical form of the instrument. His idea was to start out with already perfectly-tuned glasses, doing away with the water tuning altogether. He thought to remove the stem and base from each glass, then slide them one by one, progressing from the lowest note to the highest, on a steel shaft. With a hole through the center of each glass they could all be made to nestle quite comfortably within each other, close but not touching, while the spindle could then be safely set into rotation, the glasses spinning securely along with it. Then the player would only need to touch the glass rims as they passed under wet fingers, and the instrument could be played much more like a piano, with chords and faster musical passages being much easier to achieve than having to awkwardly coordinate turning the finger around the rim of each separate glass.

Glass blowing was much more common in Franklin's time than it is now, and Franklin put many glassblowers to work on his ambitious project. For every one hundred glasses that were blown, only one ended up being properly sized, tuned and usable, so the completion of an entire four-octave glass harmonica was an arduous task. But finally, in 1761, the goal was accomplished, all 48 notes, two octaves below and two octaves above middle C. In honor of the Italian word for harmony Franklin named his invention the "armonica", or as it later came to be known, the "glass harmonica".

This instrument enjoyed a very wide popularity in Europe for about forty years. Our research indicates that there was a large glass harmonica factory with over 100 full time employees, building hundreds of instruments. Today only a few of these exist in museums around the world. Further research has also turned up over 300 original compositions for the glass harmonica, by such great 18th century masters as Mozart and Beethoven, as well as by many lesser known composers.

After about forty years the instrument suddenly lost its popularity; it quite literally vanished from public view, presumably (from what we can gather) because people came to fear its powers. They believed it caused insanity, nervous disorders, convulsions in dogs and cats, marital disputes, and even woke people from the dead. It was actually banned by police in some German communities. People so feared to touch it that a keyboard form of the instrument was devised, where by striking a key a spring would activate a wooden arm covered with a little finger of wet leather which would reach out and make contact with the glass rim, producing a note in the same manner as direct contact with the natural finger could produce one.

To what might one attribute all this craziness? One theory is that the lead in the type of glass used at the time would leach through the fingertips, into the bloodstream, and cause nervous damage. Another idea revolved around the distrust and suspicion people had of Dr. Franz Anton Mesmer, the psychologist who brought us the word "mesmerize." He was a great lover of the glass harmonica and in fact he used it in conjunction with his "animal magnetism" healing cures, to induce deeper states of hypnosis in his patients. Perhaps this use of the instrument was enough to give it a very bad reputation, encouraging people to fear it and think it evil.

Regardless of all this controversy, during its heyday the glass harmonica was the talk of the town, said by some to be more popular than the violin. Its sounds have been described as ethereal, haunting, ghostlike, mystical, angelic, coming from nowhere, pervading everywhere... Two of the leading virtuosos of the day were a blind woman and friend of Mozart's by the name of Marrianne Kirchgaessner and the niece of Ben Franklin, Marianne Davies.

Now, finally, after over 150 years of obscurity, this fascinating instrument is back in production, being built by a master glassblower from Waltham, Massachusetts, Mr. Gerhard Finkenbeiner. Finkenbeiner saw one of the original instruments in a museum in Paris in 1960 and carried the idea and the desire to build one himself in head head for many years. He completed the first prototype in 1980, based on the original Franklin blueprints, but also incorporating a few changes based on modern glass manufacturing capabilities and technologies.

First, the foot-powered treadle used to turn the original spindle has been replaced by a silent 110 volt electric motor. When Franklin designed the foot treadle mechanism used on his original instrument he was faced with several problems.

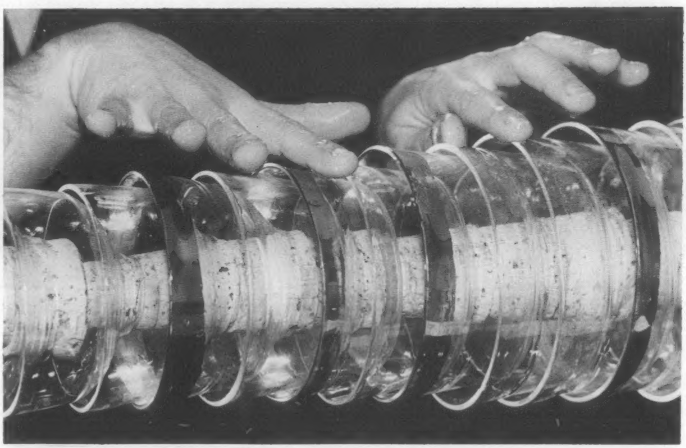
One was the slow speed (30-50 rpm) that was necessary to get a good response on the middle and lower glasses. It is difficult to operate a foot pedal at low speed. Franklin installed a heavy flywheel on the direct drive spindle (25 pounds of lead) and solved the problem this way. Later instruments used speed reduction mechanisms, allowing the spindle to turn more slowly than the cycle of foot pedaling. The ultimate solution was found on some later instruments made around the turn of the century. The rather large flywheel was tacked horizontally under the spindle in a sort of double-bottomed suitcase where it could hardly be noticed. A clever leather belt arrangement drove the flywheel at high speed while the spindle rotated slowly.

Today Finkenbeiner uses a constant-speed motor which uses a voltage feedback regulation. When set at 50 rpm it will hold that speed quite well, even if the load on the glasses is increased, for example, during the playing of chords, crescendos, etc. The Dayton DC motor controller model 5X412A used with the Dayton DC motor 42142 offers reasonable quietness while doing the job. The player controls rotational speed by having a foot pedal attached to the knob on the speed control.

For a second modern innovation, the type of glass used now is far superior to the old soda lime glass previously used. The individual cups (or bells) are fashioned from 100% pure silica, or semiconductor grade fused quartz, the highest quality glass known to man. This is the type of glass Finkenbeiner uses daily in his shop for doing his more traditional work as a scientific glassblower.

Finkenbeiner begins the manufacturing process by mounting long tubes of quartz glass on his glass lathe. For the lower notes he will use tubes of larger diameters; for the higher notes he will use tubes of smaller diameters. Then, by a combination of blowing and turning, he tools the molten glass at temperatures around 2000°C, fabricating a series of

Above: GLASS HARMONICA BY GERHARD FINKENBEINER. Below: AS PLAYED BY VERA MEYER (Photos by Tom R. Chambers & Betsy Cullen).



UNASSEMBLED PARTS FOR THE GLASS HARMONICA (from a kit, available for purchase).

Photo by Betsy Cullen

elongated spheres along the entire length of the cylinder. Each of these spheres will later be sliced in half to produce two unrefined glass harmonica bells. With the aid of a musical stroboscope Finkenbeiner then classifies each cup according to the note it is closest to in pitch. Holding each glass loosely in his hand he raps it sharply with a stick to make it ring in its particular frequency. It would take years to complete a single harmonica if one went about trying to make one note at a time precisely the right shape and pitch. To circumvent this problem Finkenbeiner uses the approach of making hundreds of cups of all shapes, sizes, and tonal qualities at random. In this way he acquires an unrefined supply of many potential middle C's of all different sizes, many C#'s, many D's, etc. It took him about one year to produce a stock of 500 cups at this rough stage of completion.

When his supply of cups is large enough Finkenbeiner turns to the real precision work, that of fine tuning each cup to exact concert pitch. He does this in the following way: if the cup is slightly flat he grinds away at the base of it to create less mass; this makes the pitch higher. If however the note is too sharp he uses a different process, acid etching. He immerses the cup in hydroflouric acid solution for a precise amount of time. The acid eats away the glass, thinning the walls and thus lowering its pitch. If for example he were to take a middle C cup with a base diameter of 4" and immerse it in a 50% hydroflouric acid solution for 20 minutes, two thousandths of an inch (.002") of wall thickness would be removed, and the former note C will now be B, one half step lower. This explains why a glass blower has a difficult time producing a given note <u>and</u> a given diameter. A glassblower can control wall thickness within $\pm$ ten thousandths of an inch. Therefore the note he or she is making might be five halftones higher or lower than the one needed in order to fit perfectly as the next note in sequence on the glass harmonica currently under construction. Once again, it would be an impossible task to complete a harmonica by making one note at a time.

With his large supply of tuned notes of various sizes and diameters the last step is now easy. Finkenbeiner has only to select the best fit out of his large selection of each note when choosing the cups to slide one by one, side by side, on the spindle. To mount the cups securely on the spindle he has devised a method of using corks as the ideal material to use as a buffer between the bare metal rod and the glass. The corks get holes drilled through their centers and are slid on the spindle first; the cups come sliding directly on top of the corks. The corks are cut to fit just perfectly to keep the cups both tightly seated in correct position on the spindle, wedged in, and also to keep them distinctly separated. It is critical that the glasses not touch each other, to ensure that the vibration of each individual glass is free and unrestricted, with unimpeded rotation through the air. The trick is to get the glasses very close to each other, perhaps with a $\frac{1}{2}$" space between rims, but still not touching.

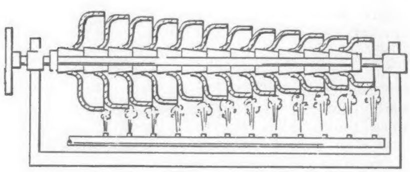
There is one other problem that occasionally needs attention. Some cups, when vibrating, produce two distinctly different notes at the same time, due to slightly uneven walls. The two notes may be apart by the interval of an even halftone. Looking at the large end of the cup there will appear a precise pattern of frequencies. At 0, 90, 180 and 270 degrees is one note, then at 45, 135, 225 and 315 degrees is the other one. Today's modern methods using microphones and stroboscopes can determine the exact location of both frequency peaks on the cup. The use of precise spot grinding techniques will bring the cups down to only the lower note, without altering the lower pitch at all.

For years Finkenbeiner has experienced sounds

with all possible types of glass. In judging their sound in the harmonica he finds quartz glass by far the best, being of superior resonance, followed by lead glass (crystal), then soda-lime glass (regular drinking glass glass), and finally borosilicate (pyrex).

With the glass harmonica completed for the specific range of notes he desires, Finkenbeiner now adds one last touch of practicality combined with beauty. Using liquid gold he paints a gold band along the inside border of just the sharps and flats, to mark them just as ebony wood marks the black keys on the piano. Baking these cups at high temperature will permanently fix the gold to the glass, marking these notes for life, thus serving as a visual aid to orient the player with reference points in the musical scale.

As an optional feature Finkenbeiner offers a misting apparatus, a length of tubing with small holes placed at regular intervals, which is positioned underneath the glasses. A small generator pumps steam through the tube, and it gets forcibly ejected through the holes, spraying the glasses with regular doses of steam to keep them moist. One can do without this option by simply keeping two water bowls near the instrument, dipping the fingers whenever more wetness is required.



The glass harmonica is an idiophonic instrument. It is different from most other idiophones in that it is not struck percussively, but rather rubbed. The vibration is not set in motion by a blow but rather by the same principle we see acting in a violin, i.e., the sticking and slipping of the bow on the strings. The key element in this stick/slip principle is friction. In a violin bow we use rosin for grip -- on a glass harmonica we use water on the well-washed finger. If the fingers are not "squeaky clean", having traces of body oils on them, the glasses will feel only slippery under the touch, there will be no firm grip established in contact, and there will consequently be no sound produced. With fiction at work the finger is constantly sticking to the glass and only slipping on it when the momentum becomes too great. This repeated action causes the vibration of the glass and the resultant note in the frequency of that vibration.

The optimal speed of rotation is around 50 rpm. If the speed is too slow an even and solid tone will not be produced; if it is too fast the glasses will revolve too quickly to allow the finger to develop any grip at all, and there will be only slipping. Since the higher notes are smaller in diameter they could use a slightly faster rate of rotation than the lower notes to allow the rims to pass under the fingertips at the equivalent rate of speed.

So far as I know no one has yet studied the acoustics of the glass harmonica from a scientific standpoint. People have told me the sound seems very pure but the harmonics quite complex. It would be interesting to play a glass harmonica into an acoustic spectrum analyzer to see what interesting things might be revealed by this wave analysis. The sound of this instrument is altogether otherworldly, having a somewhat theramin-like quality. One has to hear it to believe it!

As one of the few players of this instrument I have given over 200 concerts within the last three years. My audience has ranged from scientists to New-Agers, from the very young to the very old, from classical music buffs to synthesized music lovers. I do not like to restrict my repertoire to only those classical works composed specifically for the instrument; I play what I like and also what I feel sounds good on the instrument, from classical to popular, folk, Irish and Japanese. I enjoy appealing to a wide range of musical tastes. It will not be long before contemporary composers find out about the glass harmonica and begin writing new works for it. Several people have already approached me about this possibility. We welcome anyone interested to the budding new world of glass music. In fact, we are currently in the process of founding a non-profit organization for glass music enthusiasts, GMI, or Glass Music International. We intend to revive the lost art of glass music. For further information or questions please write Vera E. Meyer, c/o Gerhard Finkenbeiner, 33 Rumford Ave., Waltham, MA 02154.

Recordings of the glass harmonica are now available. They are:

VERA MEYER PRESENTS THE GLASS HARMONICA -- $6 including postage. One hour of exclusively glass harmonica music, including classical, popular and folk music.

THE ART OF THE GLASS HARMONICA by Kenneth Piotrowski -- $10 including postage. Kenneth Piotrowski is the world's leading virtuoso on the instrument. He performs only original classical glass harmonica compositions, both solo and accompanied by soprano voice and string quartet.

GERHARD FINKENBEINER PRESENTS GLASS MUSIC -- $5 including postage. Gerhard Finkenbeiner, the builder of the glass harmonica, plays his own original music combining glass harmonica, synthesizer, and another of his musical creations, the quartz glass carillon.

Any questions about the glass harmonica, concert bookings, or orders for tapes should be directed to Gerhard Finkenbeiner, 33 Rumford Ave., Waltham, MA 02154; phone (617) 899-3138.

SIGNAL PROCESSING WITH MICROCOMPUTERS

Does it seem surprising to see an article on computer music systems in Experimental Musical Instruments? In the past EMI's orientation has always leaned away from purely electronic devices. Two considerations have led to the appearance of this article: First, signal processing systems work with natural sounds sources as raw material. This is especially true of sampling systems. Second, a great many people who are exploring new acoustic sound sources are applying electronic sound modification to their work.

At the heart of current sound processing technology is our relatively new ability to digitize sound -- that is, to represent the form of a sound wave by a series of numbers. This ability has huge implications for storage, manipulation and reproduction of sound. When sound information is converted to numerical data, it becomes amenable to mathematical manipulation by computers, with a degree of flexibility far surpassing anything previously possible. Furthermore, numerical data can be accurately reproduced over and over without the gradual degradation of the signal content that occurs with other media.

In this article author David Courtney gives an overview of the basics of digital signal processing using a microcomputer. He gives some history and background on sound modification, describes a simple hardware setup, and, in the heart of the article, describes the workings of some of the most musically useful "effects" -- that is, some of the ways in which the signal can be modified to produce particular aural results. Many of these are the effects that are commercially available in the form of dedicated hardware (wah-wah pedals, delay units and the like). The great advantage of doing it yourself with software instead, of course, is unlimited flexibility -- to say nothing of fun.

Digital sound has been the subject of some controversy, in part because of extravagant claims that were made in connection with the appearance of the current generation of compact discs. But CDs are just one application of digital sound technology, with their own operating systems and standard sampling rates. Synthesizers, sampling keyboards and other ready-made hardware likewise have their built in systems. With home computers, the user can design his own, made to fit his needs, within the limits of the speed and flexibility of the computer.

The creation of a computer signal processing set up from scratch is a complex task with a million details and variables. Specific procedures are dependent to a great degree on the particular hardware and software the user works with. Accordingly, this article provides an overview rather than specifics. For those wishing to know more, further reading is recommended at the end. And, of course, there is no substitute for hands-on experience and exploration.

David Courtney is owner/president of Sur Sangeet Services, a small company dealing with a variety of music related activities including multitrack recording, analog/digital sound processing, and high speed cassette duplication. He has studied Indian classical music and is currently music section head for the Anjali School of Indian Dance and Music.

NEW SOUNDS FROM OLD SOURCES

Musical Signal Processing with a Microcomputer

by David Courtney

History of Sound Modification

Men have played musical instruments right from the dawn of civilization. At the same time there has been a continuous quest to modify the sounds of existing instruments. Some classic acoustic means toward modifying the sound have been rather ingenious. Let us take the "pan" on a dobro guitar, or the incorporation of a flat bridge on the Indian sitar for two examples of the drastic changes in the timbre which can be produced. The mute for horns or other instruments is another example. A kazoo is a good example of an acoustic means of modification for the sound of the human voice.

The acoustic means of sound modification have been far from perfect. It is true that they are usually portable, inexpensive, and highly complex as to the physics of their operation, but they have one serious drawback. That drawback is inflexibility.

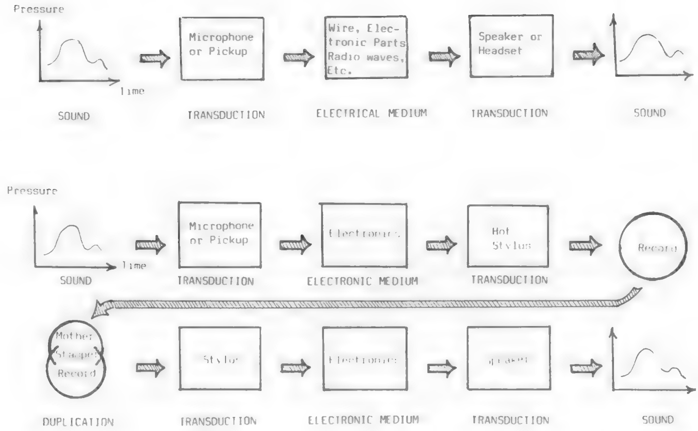
The inflexibility of acoustic techniques had to be accepted as a fact of life for many years. The turn of the century however, saw the seeds being planted for a totally different technique. The pioneering work of Bell, Edison and numerous others was giving a practical basis to the concept of transduction. Transduction is the transference of the information in one medium into another. The early researchers were taking information about changing air pressure (i.e. sound) and converting it to an analogous changing electrical signal and then back to sound again. We tend to look back and be unimpressed by their level of technical achievement but we should realize the philosophical implications of their works. They brought to mankind the concept that information has an existence separate from the medium of that information. The ramifications of that concept continue to haunt us even today.

The 20s and 30s saw the creation of a massive electro-acoustic system for sound reproduction. This system in its simplest form consisted of three sections, as shown in the diagram above right. Often it was more complicated, as in the lower diagram.

The problem was that each section was prone to certain distortions. To be more specific, there were certain non-linearities in response. The

Figure 1

THREE BASIC SECTIONS OF THE ELECTRO-ACOUSTIC SYSTEM FOR CONVERSION OF SOUND INFORMATION TO ANOTHER MEDIUM AND BACK INTO SOUND. OFTEN THE SYSTEM IS MORE COMPLEX, AS IN THE LOWER DIAGRAM.



bottom line was that what the listener finally heard was not what the player had played! For many years these non-linearities were viewed purely as a nuisance. When they could not be eliminated, they would at least be worked around.

The process of working around the inherent weaknesses of this system produced results which were both immediate and quite profound. Early microphones had poor sensitivity and very weak high frequency response so emphasis was placed on instruments that had powerful overtonal characteristics and were very loud. Trumpets and saxophones were an obvious choice given the circumstances. When strings were required it was necessary to place them as close as possible to the microphone. The speakers could not produce bass effectively, so string bass, contrabassoon, etc., were definitely out. One can go so far as to say that the "big band" sound of the 30s and 40s was due in part to the limitations of recording technology and the radio broadcast system.

The 50s and 60s ushered in a whole new era. This period saw the second generation to be raised on electronically filtered music and the first to scarcely even know what pure acoustic music sounded like. Even supposedly "live" music was nearly always heard through the medium of amplifiers and large speakers. The improvements in fidelity allowed the transmitted recorded music to sound more like the original than ever before, but since few people knew what the original sounded like, natural sound became less important. The musical philosophy had shifted away from the elimination of non-linearities to the judicious use of controlled non-linearities. These controlled non-linearities are more commonly referred to as "effects".

All three blocks of our system (transducer, transmission, speaker) were originally used for particular effects. The speaker side saw the first fuzz as nothing more than a speaker with holes in it. The granddaddy of the present flangers were two speakers with fan blades revolving over a port in the top. But the most ingenious effects of all were purely electronic. Wah-wahs, fuzz boxes, and a host of other "black boxes" were the order of the day.

The 70s and early 80s saw the elimination of transducers and speakers as a source of effects and the dominance of purely electronic means. The reason was simply that the acoustic component of both ends made them quite inflexible, just as purely acoustic means were inflexible. This period saw the more widespread use of effects, but did not see any substantial increase in the number of new effects. The only exception would be the time oriented effects such as delay, echo, phasing, flanging, etc. These effects were made possible by advances in MOS (metal oxide semiconductor) technology. This same technology was responsible for the introduction of a chip that would not only revolutionize music, but the entire world as we know it.

This revolutionary new chip was known as the microprocessor. The first real microprocessor was the 8080 introduced by Intel in the mid 70s. This was the vanguard of an entire line of "computer on a chip" devices, whose prime function was to process information. Even though microcomputers based on these devices have been around for more than ten years, their musical possibilities have only just now begun to be exploited. We will now look at some ways these microcomputers can be used to process the sound of natural acoustic instruments.

Fundamentals of Digital Audio

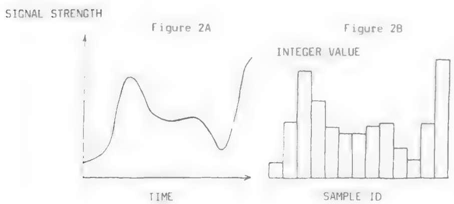
Computers are very fast, efficient number crunching machines. However, before they can be used to process music, a means must be found to convert it into numbers and back again. The conversion between music and numbers involves two

processes. The first process is transduction, whereby variations in air pressure are converted into analogous variations in an electronic signal. The second process is digitization, whereby smooth variations in the electronic signal are sampled and discrete numeric values assigned to them. The conversion from numbers back to music is simply the reverse of the previous process. Since the transduction process simply involves microphones and speakers, we need not discuss it here.

ANALOG TO DIGITAL CONVERSION: An analog to digital converter (A/D for short) is the circuit which takes a continuously varying input and generates discrete numeric samples as its output. This may be a little difficult to understand, so let us resort to some illustrations to clarify these concepts. Figure 2(a) shows a hypothetical signal. We can input this signal to our A/D converter and get numbers which correspond to the bar graph on fig 2(b). This whole process is analogous to a motion picture camera which captures moving events in a series of discrete frames.

Figure 2A

Figure 2B



One of the most important points to consider is the rate at which the input signal is to be sampled. We can see from the previous illustration that increasing the sampling rate amounts to increasing the horizontal resolution. Obviously there are practical limits to our sampling rate so let us take a look at some of the relevant factors.

We previously compared A/D conversion to a movie camera. We all know that the motion picture camera works very well with slow moving objects (people, cars, etc.), but we also know that it does not work well with fast moving objects (wheels, airplane propellers, etc.). We have all seen propellers appear to move slowly, stop, move backwards and in general exhibit action totally inconsistent with its actual motion. This same phenomenon is present in digital audio with rapidly changing input signals. If the signal exceeds a certain frequency (known as Nyquist frequency) it becomes reflected back down the audio spectrum in a most unpleasant manner. This totally unrelated signal is known as aliasing noise and it must be minimized.

There are two ways to minimize aliasing noise. The essence of the Nyquist theorem states that any frequency greater than one half the sampling rate cannot be sampled with any meaningful results. Therefore if we decide what the highest frequency we wish to track is, and set the sampling rate to at least twice that, aliasing noise will be reduced. The second method involves filtering out everything above the Nyquist frequency. The reason is simple. Many musical instruments have overtones which extend into the high audio and even the ultrasonic frequencies. When these overtones enter the A/D converter at greater than the Nyquist point, aliasing noise invariably results. A good low pass filter usually takes care of everything.

It would appear that having a very high sampling rate would solve all problems. Unfortunately that is not the case. Let us say that we want a 20 kHz response (20,000 cycles per second, or the upper limit frequency response of an ordinary stereo amplifier). According to the Nyquist theorem we would need a sampling rate of not less than 40,000 samples per second. That means that your computer must perform all operations in 25 millionths of a second! One can see that the processing options are very limited by a high sampling rate. But where there is a will there is a way, and some ingenious approaches have been worked out to produce some very interesting musical effects within the limitations.

Another consideration is the number of bits (units of information storage) to be used in the computer code which assigns digital values to the samples. We can look back at figure 2(b) and see that if we want accurate reproduction we have to think of our vertical resolution. The more steps we have on our vertical axis, the greater will be our fidelity. Each step must be assigned a unique value to identify it for processing and reproduction purposes. The number of discrete values available is linked to the number of bits in the code used. A single bit can define only two steps (0 or 1). A two bit code could define 4 steps (00, 01, 10, 11), a three bit code could define eight steps, etc. The relationship can be expressed as:

$$\text{(NUMBER OF STEPS)} = 2^{\text{(NUMBER OF BITS)}}$$

Therefore if we were using an 8 bit code, we would have 256 possible steps. In practice, an 8-bit code is usually appropriate for any of the 8-bit machines on the market, and something between 8 and 16, depending on various factors, for a 16 bit machine.

This is all well and good but audiophiles are used to thinking in terms of signal to noise rations (S/N ratio). It turns out that the maximum S/N ration can be calculated as:

$$\text{(S/N RATIO IN dB)} = 6 * \text{(NUMBER OF BITS)}$$

Therefore the most that one could expect of an 8 bit system would be a S/N ratio of 48 dB. This is not exactly "Hi Fi" but it is certainly comparable to many "black boxes" on the market today.

DIGITAL TO ANALOG CONVERSION: We have discussed some of the points relevant to A/D conversion, now we should discuss digital to analog (D/A) conversion. The D/A converter takes the binary number

and converts it to a comparable electrical signal. Its operation is usually straightforward and the only major concern is how smoothly it shifts form one sample to another. It is quite usual to find a momentary discontinuity when one sample shifts to another. The "glitches" can produce unpleasant noise. Although elaborate circuits exist to minimize these glitches, usually nothing more than a low pass filter is required.

Hardware

Let us return to the topic of sound processing. Figure 3 shows what a typical small computer-based signal processing setup might look like:

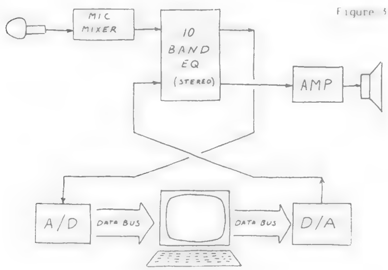


Figure 3

We see that a normal sound is picked up by the microphone. It is boosted to line level by the mic mixer and fed into a ten band equalizer. The equalizer makes a very flexible and at the same time economical filter to reduce aliasing noise. The filtered output is then fed to an A/D converter. Digitized samples are then read by the microcomputer from the data bus. The microcomputer then processes the data and again places the results on the data bus where the D/A converter reads it and generates an appropriate output. This output is then fed back to the unused channel of our equalizer to remove glitches. The deglitched signal is then amplified by an ordinary stereo amplifier and fed to the speakers where it can be heard. (That wasn't so hard, was it?)

Now we plug it in and voila! Nothing happens! The problem is that we do not have the software to support all of this hardware.

Software

The field of software has almost totally eclipsed the development of hardware in the computer industry. Testimony to this is given by the number of people who buy such and such machine solely because they want to run so and so software.

The non-availability of software for our home brew processor means that we have to write our own. I will not lie and say that it is easy. It is difficult and very time consuming. I will however give tips that will make the job easier.

The language to use is the first topic to be encountered. Most microcomputers are automatically equipped to function in Basic. Basic is convenient for many non-critical areas of our program but the main functions can not be implemented with any such high level language. We have no choice but to do the bulk of our programming in assembly code. It is possible to link certain functions together by means of the basic interpreter. The three main functions are:

1. SYSTEM & PROGRAM INITIALIZATION

We must boot our program, tell our computer what musical parameters to deal with, specify disk operations, etc.

2. PERIPHERAL SUPPORT

We must tell the computer that it has an A/D converter and a D/A converter and how to handle them.

3. MUSICAL PROGRAM

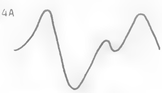
We must specify what processes are to be applied to the incoming data, and what to do with it after it is processed.

The processes in #3 above are the heart of the entire effect. What is their relationship to the audible result? To begin to understand this, let us first cover some fundamentals.

We must always keep in mind what the human ear hears. The most basic thing is that we hear frequencies but not wave forms. This is important because many electronic processes tend to affect wave forms. This is usually referred to as **time domain** (wave forms) versus **frequency domain** (frequencies and spectral content). The relationship between time domain and frequency domain is described in depth by the **Fourier Theorem**. Space limitations do not allow a complete discussion of this but the gist of it is that any repeating wave form (such as most musical signals) may be broken down into its component sine and cosine functions with their particular phase relationships. (Whew!) But we cannot hear the phase relationships of these functions, therefore we can say that any sound may be perceived as the sum of its main frequency and its harmonics (that's much simpler). There is a curious ramification of our inability to hear phase relationships. That is that indistinguishable sounds may have different wave forms. Therefore changing a wave form will usually but not always change the perceived timbre. It all depends upon whether the change in wave form produces an audible change in the spectral content (harmonic structure). This is an important point to remember because many effects work by changing the wave form, and the perceived change in sound is not necessarily proportional to the change in the wave form.

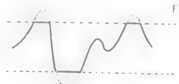
Let us now look at a classic effect which is easy to implement. This effect is **Fuzz**. Look at figure 4(a) and 4(b).

Figure 4A



UNPROCESSED INPUT SIGNAL

Figure 4B



FUZZED OUTPUT SIGNAL

Notice that the input signal is simply truncated to produce the output. This can easily be produced with a variety of hardware or software. The results of fuzz are:

1. Increased upper harmonics (especially in the odd numbered harmonics)
2. Increased sustain
3. Lower dynamic range

There is a variation upon fuzz which I call **Audio Pixelization**. This is most easily produced with computer software. Figure 5 demonstrates how it works:

Figure 5

UNPROCESSED INPUT SIGNAL PIXELIZED OUTPUT SIGNAL

What is happening here is that we take the incoming signal and, by using software, reduce the number of bits on the output. Reducing the number of bits reduces the number of discrete positions that the output can occupy. The results are:

1. Increased upper harmonics
2. Preserved percussive quality
3. Full dynamic range.

The lack of compression and full dynamic range is musically significant. We can get the harmonic content of fuzz and still retain the percussive quality of instruments like string instruments.

Delays are very popular effects which are best suited for a digital approach. Therefore our microcomputer based system should be able to handle them very well. This is indeed the case. To produce a simple delay:

1. Establish a circular buffer for the data.
2. Establish part of the program to read the A/D converter and sequentially place this in the buffer (write).
3. Establish another part of the program to sequentially read the buffer and send the data to the D/A converter.

The delay time will be directly proportional to the degree of offset between writing to the buffer and reading from the buffer.

Numerous variations upon this process can be used. For instance:

VIBRATO: Vibrato is a common effect which is characterized by a pleasant wavering of the pitch. This effect can be made from the previous arrangement with just slight modifications. We must first set the size of our buffer to some minimal value to minimize the delay. We next begin to modulate our offset value. A continual rise and fall in the offset value causes the scanning rate of the read function to waver, producing an electronic "Doppler effect". The shift in frequency will be directly proportional to the rate of change in the offset value.

PHASING (FLANGING): This can be produced from our arrangement by slowing the rate of modulation on a vibrato. We then mix the affected signal back with the original. This mix may be accomplished either with software or with hardware. The result is quite interesting. Anytime you mix a delayed signal with the original you have what is referred to as a "comb filter". A comb filter is characterized by numerous sharp variations in frequency response. The reason is simple. Some frequencies will be **in phase**, meaning they will act upon each other in an additive manner. Some frequencies will be **out of phase**, and the effect will be subtractive. The result of modulating the delay time will be a comb filter which sweeps up and down the audio spectrum.

PITCH TRANSPOSITION: This is easily produced from our original configuration. We write to our circular buffer just as we did before but we read from it at a totally different rate. This produces a shift in pitch without changing the tempo of the music. As with our vibrato, the change in pitch is proportional to our rate of change in offset value. Therefore we can simplify this by saying that the pitch change is proportional to the difference in the read/write scanning rates.

REVERBERATION: This is easily produced from our delay routine by taking a small part of our output and feeding it back to our input. This can best be accomplished with external hardware because of certain limitations in software processing.

There are a lot of functions that are not easy to perform. Multiplication, division, and exponential functions are especially difficult for computers. You are probably shaking your head in disbelief, saying to yourself "Even my $5 calculator can do these things. How can this man say it is hard for computers to do them?"

To understand the problems you must understand the relationship between a microcomputer and a microprocessor. A microcomputer is what you buy that does your word processing, plays games, and tells you that you spent more money last month than you took in. The heart of the whole machine is a single chip called the microprocessor.

Microprocessors all have a list of the things they can do. These lists are called "instruction sets". Playing games and word processing are not on these instruction sets. Instead we have such basic processes as:

1. additions
2. subtractions
3. making comparisons
4. moving information
5. branching to other parts of a program
6. Logical operations

That's it! We have no multiplications, divisions,

averages, sines or anything that we would normally associate with higher math. As a matter of fact any 6th grader can do more math than a microprocessor.

How does a computer do all of these things? Just as multiplying 220 X 50 could be reduced to successively adding the number 50, 220 times, all of our higher math functions must be reduced to successive adds, subtracts, comparisons and shifting bits around. This all takes time. Time which is measured in milliseconds. But remember we earlier discussed that we had to be finished with everything in a few 10s of microseconds. It is quite clear that this limits our options.

Things are improving with the introduction of "math co-processors". These are separate chips designed to work in tandem with the main microprocessor. They speed things up considerably but they are not yet found in all microcomputers. So for the time being we must be content with certain software tricks that allow for speedy operations.

One trick is useful anytime we need to change the level of a signal. Intuitively we would say this should be done by following the instantaneous value and multiplying by some constant. Unfortunately we do not have time to multiply. But there are some quirks in binary math that we can use. Look at the following figure:

Binary number	Decimal Equivalent
01100100	100
00110010	50
00011001	25

We can see that every time the number is shifted to the right, it amounts to dividing the number by two. Every time a number is shifted left, it is equivalent to multiplying by two. As for the audible result, it turns out that doubling or halving any number amounts to increasing or decreasing the sound level by 6 dB (note: +10 dB corresponds roughly to doubling perceived volume; -10 dB to halving it). Therefore, anytime amplification or attenuation in discrete 6 dB steps is acceptable, we can do it simply by shifting the original value.

I must inject one word of warning here. Many computers use coding systems in which the extreme left digit functions as the sign. If this is the case, it sabotages the bit shifting trick, and it won't work.

TREMOLO: This is one effect that would lend itself well to bit shifting. Tremolo is the wavering of the amplitude of the musical signal. To produce it we simply have to read the A/D converter, alternately shift left and right, and send the resulting values to the D/A converter.

DIGITAL SAMPLING

The ability to digitize sound for manipulation by microcomputers has raised yet another possibility, and it is possibly the most exciting thing to hit the musical scene in years. In the process known as sampling, we can lodge a natural sound (in digital form) in the computer and play it back every time a key is pressed. There are a number of keyboards on the market which utilize this technique. The software to drive such a device is actually not one but two different programs -- one to acquire the sample and another to play it. The first program might go something like this:

1. Input a record from A/D converter to a buffer
2. Determine whether we have an optimum s/n ratio (i.e., have we used a full excursion of our possible steps)
3. Edit the silent leading edges
4. Mark and store the record to disk for later use.

The playback performance would go something like:

1. Select the desired sample from disk and lodge it in memory
2. Check for keyboard closure (i.e., keys pressed)
3. If closure is detected determine the indicated scanning rate (slow scanning rates = low pitches; fast scanning rates = high pitches)
4. Output the record to D/A converter at the determined rate.

The beauty of this whole approach is that it allows a musician to spontaneously perform and have at his command the natural sound of a number of different instruments. The sampled sound need not even be a musical instrument. It could be any sound that has a well-defined pitch. One could just as easily use the sound of barking dogs or glass breaking.

Conclusion

The greatest strength of computer based systems lies in their flexibility. Such a system may seem complicated, yet it is presently within the reach of most people. Just in the time I have been writing this article a half a dozen sampling packages have hit the market, all of which offer the hardware to facilitate any of the processes we have described. Though they are presently aimed at the upper levels of the microcomputer market, we will see drops in the cost and increases in the capabilities, all of which offer exciting possibilities for the performing artist.

If you are interested in pursuing this further I can recommend reading:

Musical Applications of Microprocessors, by Hal Chamberlin, 1985, Hayden Book Co.

Foundations of Computer Music, by Curtis Roads and John Strawn, MIT Press

Electronic and Computer Music, by Peter Manning, Oxford University Press, 1985.

Experimental Musical Instruments regularly reports on organizations and periodicals of potential interest to its readers. This issue's column is on the Just Intonation Network. Since the subject of just intonation crops up frequently in these pages, and since it is not universally understood, we follow the information on the Network with some introductory notes on just intonation itself, its history and the current growing interest in it.

THE JUST INTONATION NETWORK

As most of EMI's readers will be aware, more and more musicians have been exploring just intonation systems in recent years. The resources for working with just intonation, for learning about it and communicating about it, however, have been scanty and not always readily at hand. In 1984, in response to this situation, several justly-oriented musicians got together to create the Just Intonation Network. The prime movers were Carola Anderson, David Doty and Henry Rosenthal, who together form the core of **Other Music**, a group of composers and performers based in San Francisco.

The network describes itself and its purposes in one of its flyers as follows:

The Just Intonation Network was founded by a group of Bay Area composers for the purpose of encouraging communication and mutual support among composers, musicians and instrument builders exploring this crucial musical frontier. Although a growing number of musicians are being attracted to the powerful musical resources offered by Just Intonation, there has been no central source for information about this evolving field. As a result, composers in different geographical areas who have similar ideas and concerns remain unaware of one another; and novices are often required to reinvent the wheel, for lack of access to primary sources. The Just Intonation Network was created to remedy these conditions.

To fulfill these purposes, the network has initiated several activities. The most prominent has been the publication of the journal, **1/1**. Volume I #1 of **1/1** appeared in the early months of 1985, and the journal has been appearing on a quarterly basis since. Articles dealing with various aspects of just intonation are written by a small staff and members of the network at large. Some of the pieces are fairly abstract or technical, though rarely oppressively formal. The merits of various just systems are often explored, along with the mathematics underlying particular systems and, frequently, charts of frequencies or ratios. A lot of articles look at broader questions relating to just intonation in general, such as the matter of notation for specialized tuning systems, and the equally imperative matter of establishing a uniform vocabulary for discussing just systems. Many articles address practical matters that arise in practicing just intonation, such as new instrumental resources, use of standard instruments and electronics, keyboard configurations and the like. Regular features include a tutorial in just intonational theory, and "Network News", reporting on goings on within the network.

A sampling of specific articles that have appeared recently includes: "Toward Standard Definitions", pointing out some of the problems and inconsistencies in terminology used to discuss just intonation and suggesting some solutions; "A Justly-Tuned Guitar", describing a refretted classical guitar; "Tuning and Transposition on the Prophet 5", describing some ways to get the most out of one of the few synthesizers not locked into equal temperament; "The Subharmonic Question", a discussion of the controversial theoretical inversion of the harmonic series that has served, ex post facto, as a model for some important harmonic constructs; and the self explanatory "A European Perspective on Partch", "Tuning the MacIntosh", and "Just Utilities Software".

The Just Intonation Network also sponsors concerts and discussion groups and lectures. Their first concert presented Pierre-Jean Croset, whose acrylic stringed instrument the Lyra was featured in an earlier issue of EMI. Topics featured in the discussions have included "Instrumental Resources for Just Intonation", "The Psychological and Physiological Basis for Just Intonation", "Just Intonation vs. the Limits of Human Perception", and a session spent listening to tapes of music composed by network members.

The network maintains an archive of manuscripts, scores and tapes pertaining to just intonation. Most of these have been contributed by members. Since no other such collection exists in the world, and so little material on just intonation is available through other sources, the network organizers regard this as one of the most important facets of their work. The network also has served as a clearing house for some commercially available materials, such as Ralph David Hill's fine cassette tutorial and demonstration tape, **The Sounds of Just Intonation**. Issue #14 of Tellus, the audio cassette magazine from New York City, was edited by the Just Intonation Network. The tape is a network compilation including previously unreleased works by Harry Partch, and Lou Harrison, among many others.

One of the subtler achievements of the Just Intonation Network so far has been to provide a sense of just what members of the new generation of microtonalists are doing. Network surveys and discussions with network members have revealed that most of the membership is working with electronics -- computers and synthesizers. Perhaps this is not surprising: microchips have opened some doors in a big way for just musicians, and can be expected to open them still wider as time passes. For Western musicians with an ingrained inclination to create multi-tiered harmonic structures (for which, in just tunings, complexities mount to an impractical degree very quickly), computers can work wonders. They also can make a very good tutor/disciplinarian when it comes to freeing musicians from an ear tuned, through a lifetime of exposure, to equal temperament.

Another general trend that has become apparent is that the majority of just musicians, following

Partch, are primarily concerned with the creation and exploration of new just systems. Not many seem to be concerned with, say, the broader application of traditional systems or the establishment of a particular new standard tuning system. Non-Western scales are definitely a subject of interest, but more as a source of inspiration than as an end in themselves. The tuning systems in which just composers compose seem to be as personal as a jazz singer's characteristic turn of phrase. The emphasis of the network and its membership, accordingly, is on exploration.

Perhaps the most important development taking place in this coming together of just musicians is the movement towards development of a rational music theory. Musicians working in just intonation are forced to look at harmonic relationships differently from what students learn in conventional theory classes. Thinking of intervals in terms of frequency ratios makes for a very precise language and some very enlightening theoretical approaches -- potentially far superior to standard theory. Harry Partch laid the foundations for the flexible yet highly disciplined rational model for musical analysis some years ago; with the Just Intonation Network in place now, we have the vessel for the ferment that will ensure its continued development.

Membership in the Just Intonation Network is open to anyone with an interest in the subject. The cost is $15 per year in the U.S., $17.50 elsewhere. It includes a subscription to 1/1, free admission to concerts and many other activities of the network, access to the archives, and reduced prices on other publications and recordings distributed by the network. The Tellus cassette mentioned above is available from the network for $7 to non-members, $6 to members. For further information contact the Just Intonation Network at 535 Stevenson Street, San Francisco CA 94103; (415) 864-8123.

SOME INTRODUCTORY WORDS ON JUST INTONATION
for anyone who has been saying
"What is all this talk about, anyway?"

Between any two notes an octave apart, there is a continuum of pitch. In taking this continuum of tonal possibility and making music from it, we humans seem nearly always to have an inclination to make it discontinuous, selecting a small number of discrete pitches to use as a tonal vocabulary. Different cultures at different times have selected different sets of pitches -- which we might as well call scales -- to work with. Why do people chose one scale rather than another for their music? What are the cogent factors in the ear's response to relationships of pitch that go into this process? This is a fuzzy sort of question, lying as it does in an obscure domain somewhere between acoustics, physiology and (the real killer) psychology. But most analyses suggest that the ear tends to hear pitch relationships largely in terms of ratios of rates of vibration, and that the more musically-meaningful ratios tend to be the simpler ones. This rule holds true fairly consistently on a cross-cultural basis.

Intonational systems which, in keeping with this mode of musical hearing, use intervals based upon selected frequency ratios are called just. People will object here: by this definition, any interval or scale system may be viewed as just, since for any two pitches there will be some ratio between their frequencies. This is true (with the obscure exception of frequency relationships which are, in the mathematical sense, irrational), so we should further refine our definition to say that in just intonations, there is a general preference for intervals accurately tuned to relatively simple ratios. Simple ratios can be defined as ratios containing no large prime numbers.

From this definition it becomes clear that just intonation is not a single scale system. Given the infinity of possible frequency ratios, an infinity of such systems are possible. A great many of them have been realized in the course of human musical endeavor. Most early European music was tuned to one just scale or another, and a great deal of non-Western music always has been and still is. But of the music happening in Western society today, relatively little is.

Starting in the late renaissance and early baroque, certain inconveniences inherent in just systems stood increasingly in the way of some important directions of European musical development. Most prominent among these was difficulty in modulating. In moving from one key to another in just scales, it usually turns out that many or most of the pitches of the new key do not coincide with any of those of the old key. This leaves us with the choice of either adding what quickly becomes an unmanageably large number of different pitches to the scale system, or accepting some very out of tune notes in the new key and sacrificing the integrity of the just tuning.

To get around these problems, European composers and theoreticians began developing **tempered** tuning systems. In such systems, certain notes were deliberately detuned a small amount to find compromise pitches that could serve in several keys. By this means the number of pitches per octave was usually kept to twelve, and it became possible to play in several keys without retuning, all with a degree of out-of-tune-ness small enough to be considered acceptable.

Early in the 18th century, one tempered system came to the fore. Indeed, it was the most tempered of the tempered systems around, but its advantages were great. What's more, it was very much in keeping with some broad trends in Western thought which have been with us ever since and have done much to shape our world. The system was twelve-tone equal temperament, in which the octave is divided into twelve equal degrees. The radical nature of this idea (cognoscenti may see a pun there) can be seen in the fact that, in order to do this, the rational thinking (as in "ratios") that had always lain at the heart of the music had to be completely discarded, and an entirely new mathematic substituted.

Given that etiology, twelve tone equal temperament happens to do a surprisingly good job of

approximating the just intervals most important in Western music. That's why it was chosen and accepted. And the flexibility, manipulability and overall convenience of a fully modular system, symmetrical every which way, proved undeniable and irresistible. But let there be no mistake about it: twelve equal and the just intervals which it approximates are not the same. The difference in sound is quite audible, and the philosophical difference inherent in the creation of a modular system is definitely not trivial.

Since that time -- for the last two hundred years and more -- the vast majority of Western music has been written and performed within the twelve-tone equally-tempered scale. Of course there are some very noteworthy exceptions to this rule. There have been renegade individual composers, and there have been whole musical styles or, on a smaller scale, stylistic devices which have stood in defiance of the preponderant tide. (Most notable among these are the blue tonalities, as long as they are realized on instruments which don't force them into twelve-equal.) But on the whole, so complete is the predominance of the equal tempered scale today that most classically-trained Western musicians, it can safely be said, never contemplate the possibility of stepping out of it. And perhaps more to the point for EMI's readers, most of the standard musical instruments are locked into it.

In recent years the number of people interested in exploring the tonal possibilities outside of 12-equal, including both just systems and alternative tempered systems, has increased considerably. Much of the credit for this must go to Harry Partch, who was, for quite some time around the middle of this century, one of very few voices crying in an equally tempered wilderness. Partch created a lot of theory, built a lot of new instruments, and rattled a lot of cages. Perversely, his music has had much less impact than his words. But while the seeds he planted may have been slow to germinate, germinated they have.

Today intonational exploration has become a force to be reckoned with in the academic arena, with several organizations and publications devoted to it, and others having at least to sit up and take note. The actual musical impact of the movement seems so far to have lagged behind the intellectual impact, but not for lack of fine music. Music in any number of deliberately created just scales or alternative temperaments is being written and performed by a varied and growing lot of musicians. Among the organizations and publications dedicated to non-twelve intonational systems now are the Interval Foundation (discussed in EMI Vol. I #5) with its publication, **Interval Magazine**; **Xenharmonikon**, an informal but highly intellectual journal newly revived and edited by Daniel Wolf in Middletown, CT; **The Xenharmonic Bulletin**, published by Ivor Darreg (no longer currently in production); **Pitch: For the International Microtonalist**, edited by Johnny Reinhard in New York; and the subject of the accompanying article and the the only membership organization in the group, the Just Intonation Network.

RECORDINGS

PARALLEL GALAXY
Emmet Chapman

An LP record featuring Chapman on the Stick, with vocalist Josh Hannah, drummer Bruce Gary and harmonica player Dan Chapman. Available for $9 from Backyard Records, 8320 Yucca Trail, Los Angeles, CA 90046.

This is the first record put out by Emmett Chapman, inventor of the Stick.

The Stick is a fretted string instrument with electric guitar pickups. It has no body to speak of (electric amplification eliminates the need for a soundbox), but is mostly neck -- literally a fretted stick. It is 3¼ inches wide, and 3½ feet long, with tuning machines at one end, bridges and an anchoring mechanism for the strings at the other. There are five strings for melody and five more for bass and chordal accompaniment, and the range reaches to five-plus octaves.

The special feature of the Stick is the fact that the strings are not plucked or bowed, but sounded by the technique guitarists usually call "hammering on": tapping them with the fingers to bring them down firmly against the fretboard. The resulting abrupt contact with the fret is enough to set them into vibration.

The hammering on technique has long been used with acoustic fretted instruments as an intermittent alternative to the usual plucking. It tends to lack volume, though, and becomes awkward if used for more than brief passages. With the benefits of amplification, the need to really hammer the string to generate adequate vibrating energy is reduced, and more extensive use of the technique becomes more feasible. Many people will remember Jimi Hendrix dazzling everybody with his one-handed playing (long passages with no plucking) back in those dramatic Woodstock days. More recently, in what has been hailed as a major innovation, electric guitarist Stanley Jordan has dispensed with plucking altogether and taken to hammering on with both hands, still using a standard instrument. The keyboard-like fluidity of his often contrapuntal, jazz-tinged meanderings has opened a lot of eyes.

With the Stick -- which, incidentally, predated the public's fascination with Stanley Jordan and his technique -- we have an instrument specifically designed to take full advantage of the possibilities inherent in the two-handed hammer on technique. These possibilities are considerable. All that is required to produce a note is tapping with one finger. Eight fingers are independently available for tapping. For the ease with which a great many notes can conceivably be produced, this arrangement approaches that of a keyboard -- assuming that the player can develop a type of psychomotor coordination and digital independence which turns out to be quite foreign to the requirements of conventional fretted instrument playing. Developing the technique has proven to be quite a challenge for those who undertake the Stick seriously, but then so is contrapuntal

keyboard technique. There have been many testimonials to the effect that the result is worth the effort.

Chapman made the first prototypes for the Stick in the early 1970s, arriving at what was to become the standard stringing, tuning and design around 1973. With the instrument being produced in ever-larger numbers, the most recent innovation has been a switch from the ironwood body that had been standard to an extremely rigid body of reinforced injection molded polycarbonate resin.

Virtually from the start Chapman has aimed for widespread popularity for the instrument, and taken a commercial approach to its manufacture. He has taken out patents, formed a corporation entitled Stick Enterprises, and built up a successful business. The musical possibilities of the instrument have been enticing enough to attract some well known musicians, and they have done much to bring it recognition. Most prominent among them -- at least in Stick Enterprises' literature -- have been Tony Levin, who has played with King Crimson and Peter Gabriel, and Alphonso Johnson, who has played with Santana and Weather Report.

Parallel Galaxy is Chapman's first commercially-available recording. Chapman has been performing on the Stick since its creation, gaining a reputation as (appropriately) its most advanced practitioner. This recording, accordingly, has long been awaited.

Perhaps we should say from the start that the record's success from a purely musical point of view is inconsistent. But as a demonstration of the Stick, its sound, its capabilities, and its viability as a musical instrument in a variety of contexts, it can safely be called a success.

The style of the music varies from cut to cut. Much of it might be called adult contemporary, with the mix of pop and a light jazz feeling that the term suggests. In other places a folky feeling comes to the fore, and other parts might be called new age. From these observations it should become apparent that the instrument lends itself nicely to existing styles, rather than demanding made-to-order music or (on this recording at least) breaking new ground.

Chapman's sound is in some ways reminiscent of a harpsichord, with a very precise, keyboard-like articulation. (The analogy to the clavichord applies to the means of string excitation but is less appropriate in describing the sound.) With the use of electric guitar-style pickups and the absence of body resonance from the instrument, a familiar element of electric guitar sound is present, but masked by the very different articulation. Chapman makes modest use of signal processing, never disguising the fundamental sound of the instrument. Perhaps one of the main factors in the impression the instrument makes on this recording is simply Chapman's virtuosity -- the precision and fluidity of playing, often quite fast and fully contrapuntal. And it <u>is</u> remarkable what the instrument can do. The listener has to remind himself frequently that it is one instrument playing all those notes.

The other performers appearing on the record -- a harmonica player, a drummer and a vocalist -- do not fill out the sound (since the Stick fills as completely as a keyboard) but rather add some color. Among them, the vocalist, Josh Hanna, comes across most effectively, with an idiosyncratic musical personality that stands out even on a recording planned as a showcase for another instrument.

Chapman has also written an instruction book for the Stick, entitled **Free Hands**. This book, as well as the Stick itself plus accessories, can be purchased from the address given at the start of this article.

LARK IN THE MORNING SEARCH AND SELL SERVICES

Builders of unusual instruments looking for ways to sell their work have a possible retail outlet in **Lark in the Morning**.

Lark in the Morning is an outlet for musical instruments based in Mendocino, California. They specialize in unusual and hard-to-find musical things, mostly but not exclusively traditional, generally made by small craftsmen. Selling is through the retail store in Mendocino and by mail order. In addition to retailing instruments, the people there organize workshops, concerts, festivals and the like. The Lark in the Morning catalog is an adventure unto itself, and a separate newsletter, **The Lark's March**, contains short articles on various subjects related to traditional instruments and music as well as goings on at the store.

Lark in the Morning takes many instruments on consignment. They charge 25% of the retail selling price for their services. The person consigning the instrument to them is responsible for shipping charges to the store and return charges if necessary. The Lark in the Morning people have been around for a long time and have made it their business, with considerable success, to take risks on unusual items. Bear in mind, though, that they are interested only in the work of serious, dependable builders, producing instruments of high quality craftsmanship and design.

Also, for anyone interested in locating unusual instruments not available elsewhere, Lark in the Morning can run searches for instruments they do not have in stock. Most of the time they are glad to do this free of charge. In cases where an instrument does not have a high value or is extremely difficult to find they will run a search for a customer for an hourly fee.

For more information on these services, or to get Lark in the Morning's catalog or newsletter, call (707) 964-5569 or write Box 1176, Mendocino, CA 95460.

RECENT ARTICLES APPEARING IN OTHER PERIODICALS

Listed below are selected articles of potential interest to readers of **Experimental Musical Instruments** which have appeared recently in other publications.

BRAZIL'S WALTER SMETAK: AN INTRODUCTION, by Jon Scoville in his "Instrument Innovations" column in **Percussive Notes** Vol. 25, #1, Fall 1986 (214 West Main St., Urbana, IL 60801-0697).

Walter Smetak (now deceased), a Swiss born Brazilian, built diverse instruments which stretched not only conventions of form and design, but basic tenets of performance practice as well. Scoville introduces us to a remarkable man and some of his creations in this overview.

KINESONE I: A KINETIC SOUND SCULPTURE, by Lin Emery and Robert Morriss, in **Leonardo** (Pergamon Press, 2020 Milvia St., Suite 310, Berkeley, CA 94704).

This article describes an aeolian sculpture installed in the 5th floor outdoor courtyard of a hotel in New Orleans. Aluminum bars, sounded by tiny concealed spring-mounted hammers, are mounted marimba-style over openings in the sides of long, upright tuned aluminum resonating tubes which pivot in the wind.

NEWBAND, CARNEGIE RECITAL HALL, MAY 9, 1986, by Karen Campbell in the "Eareviews" section of **Ear**, Volume 11 #1, August/September 1986 (325 Spring St., Room 208, New York, NY 10013).

This review of a recent performance gives a lot of space to Newband's trademark instrument, Dean Drummond's Zoomoozophone, a 31-tone metallophone.

LUMINENTA 1, by Otto Pietz, in **Folk Harp Journal** #54, Fall 1986 (4718 Maychelle Dr., Anaheim, CA 92807-3040).

In this article the author describes a harp of conventional shape and dimensions which he made, all but the soundboard, of aluminum.

WHOSE HARP, by Rick Morris, in **The Lark's March**, Fall/Winter 1986/1987 (P.O. Box 1176, Mendocino, CA 95460).

Amid the usual eclectic mix of treasures in the **Lark's March** may be found this short piece on Jew's Harps and their culture, complete with bibliography and discography. **Lark's March** is the newsletter put out by Lark in the Morning, a retail store for unusual instruments.

EXPERIMENTAL VIOLIN ACOUSTICS, by Dr. George Bissinger, in **American Lutherie** #7, Fall 1986 (8222 S Park Ave, Tacoma, WA 98408).

This piece discusses some of the factors affecting violin soundboard behaviors and experimental methods for analyzing and controlling them. Some of the things that are often discussed in **Journal of the Catgut Acoustical Society** are synopsized here, most notably, the practice of top plate and bass bar tuning.

MORE ABOUT THUMB PIANOS, by Richard Selman in **The Gourd** Vol. 16 #3 (American Gourd Society, P.O. Box 274, Mt. Gilead, OH 43338).

Richard Selman provides information on several materials, both metal and non-metal, traditional and non-traditional, which can be used for keys on thumb pianos.